AMENDMENTS TO THE SPECIFICATION:

At page 8, last paragraph, starting on line 24, please change to read as follows:

The pulse positions of each of the pulse systems 25 are limited as illustrated in Fig. 26. In the algebraic codebook search, a combination of pulses for which the error power relative to the input voice is minimized at the reproduction is decided from among the combinations of pulse positions of each of the pulse systems. More specifically, with βopt as the optimum pitch gain found by the adaptive-codebook search, the output P_L of the ~~adoptive~~ adaptive codebook is multiplied βopt and the product is input to an adder 11. At the same time, the pulsed signals are input successively to the adder 11 front the algebraic codebook 8 and a pulse signal is specified that will minimize the difference between the input signal X and a reproduced signal obtained by inputting the adder output to the LPC synthesis filter 6. More specifically, first a target vector V for an algebraic codebook search is generated in accordance with the following equation using the optimum adaptive codebook output P_L and optimum pitch gain βopt obtained from the input signal x by the adaptive-codebook search:

At page 26, first paragraph, starting on line 2, please change to read as follows:

Fig. 2 is a block diagram illustrating the voice code conversion unit in which the construction of the code converters 82 [[t0]] to 85 is clarified. Components in Fig. 2 identical with those shown in Fig. 1 are designated by like reference characters. The code separator 81 separates an LSP code 1, a pitch-lag code 1, an algebraic code 1 and a gain code 1 from line data (the voice signal based upon encoding method 1) that enters from the transmission path via an input terminal #1, and inputs these codes to the code converters 82, 83, 84 and 85, respectively.

At page 64, last paragraph, starting on line 22, please change to read as follows:

With regard to odd-numbered subframes, a point in common is that the difference between integral lag Told in the previous subframe and [[pith]] pitch lag in the present subframe is quantized. With respect to the number (six) of quantization bits in the AMR method, the

number is smaller than that (five) in the G.729A method. This makes necessary the following expedient:

At page 68, paragraph 2, starting on line 6, please change to read as follows:

Next, adaptive codebook gain code I_GAINc(m,0) of the 0th subframe in the mth frame of the AMR method is input to the noise codebook gain dequantizer 85_{a2} to obtain the algebraic codebook gain dequantized value Gc. In the AMR method, interframe prediction is used in the quantization of algebraic codebook gain[[.,]] i.e. gain as V predicted from the logarithmic energy of algebraic codebook gain of the past four subframes and the correction coefficients thereof are quantized. To accomplish this, the noise codebook gain dequantizer 85_{a2}, which has a 5-bit (32-pattern) correction coefficient table the same as that of the AMR method, finds a table value gc of a correction coefficient that corresponds to the code I_GAIN1c(m,0) and outputs the dequantized value Gc = (gc' x gc) of algebraic codebook gain. It should be noted that the gain prediction method is exactly the same as the prediction method performed by the AMR-compliant decoder.